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Robust Transmission of H.264/AVC Streams Using Adaptive Group Slicing and Unequal Error Protection

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We present a novel scheme for the transmission of H.264/AVC video streams over lossy packet networks. The proposed scheme exploits the error-resilient features of H.264/AVC codec and employs Reed-Solomon codes to protect effectively the streams. A novel technique for adaptive classification of macroblocks into three slice groups is also proposed. The optimal classification of macroblocks and the optimal channel rate allocation are achieved by iterating two interdependent steps. Dynamic programming techniques are used for the channel rate allocation process in order to reduce complexity. Simulations clearly demonstrate the superiority of the proposed method over other recent algorithms for transmission of H.264/AVC streams.

1. INTRODUCTION

The demand for multimedia transmission over best effort networks, like the Internet, motivated most recent research on real-time streaming applications. However, due to the explosive growth of the volume of transmitted data and bandwidth variations, networks employing the Internet protocol (IP) exhibit packet erasures. Considering that the network is unaware of the transmitted content, we realize that packet erasures during transmission can cause significant problems in demanding applications such as video streaming. Error-resilient coding schemes like the H.264/AVC standard [1, 2] have been proposed to overcome these problems. The H.264/AVC standard supports valuable error-resilient tools to cope with erased packets, while it outperforms previous coding standards (H.263, MPEG-4). Unfortunately, these tools increase the computational complexity, which is undesirable for real-time video applications, and have a negative impact on compression efficiency. Therefore, schemes combining unequal error protection (UEP) algorithms with appropriate selection of error-resilient tools are often shown to be advantageous for transmission of H.264/AVC-coded streams, while maintaining the computational cost at reasonable level.

In a recent work [3], data partitioning of H.264/AVC and high-memory rate compatible punctured convolutional codes (RCPC) [4] were proposed for video transmission over wireless channels. RCPC codes were applied to the network adaptation layer (NAL). Data partitions were unequally protected according to their significance. A similar approach was presented in [5], which also used the data partitioning mode of H.264/AVC. The transmitted data were protected by Reed-Solomon (RS) codes applied at the video coding layer (VCL). Unequal channel rate allocation was performed using Lagrangian optimization techniques. The efficiency of H.264/AVC error-resilient tools was evaluated in [6]. Reed-Solomon codes and a feedback channel were considered for robust transmission. Robust transmission of H.263 [7] streams was examined in [8]. A packetization method of slices and an UEP algorithm for joint optimization of macroblock coding parameters and selection of FEC codes were presented.

Partial Reed-Solomon codes (PRS) were used in [9] for reliable transmission of H.264/AVC streams over packet erasure channels. The resulting scheme was able to reduce jerkiness and improve video quality. The concept of key pictures was introduced for H.264/AVC in [10]. The source encoder was appropriately modified to generate packets of unequal importance which are unequally protected. An algorithm which adaptively classifies the data packets of MPEG-2-encoded video streams into two quality of service (QoS) classes was proposed in [11]. Packet classification into priority classes was also studied in [12]. Intraframe interleaving and RS codes were used to improve error resilience.

Encoding parameters	MB_1	MB_2	···	MB_L

FIGURE 1: Structure of slices.

The scheme proposed in the present paper is based on macroblock classification and unequal error protection of H.264/AVC streams. Prior to transmission, macroblocks are classified into three slice groups by examining their contribution to video quality. Since the transmission scenarios are over packet networks, facing moderate to high packet loss rates, RS codes are used for channel protection. RS protection is selected for each slice group using a channel rate allocation algorithm based on dynamic programming techniques. To the best of our knowledge, the present method is the first utilizing the explicit mode of the H.264/AVC flexible macroblock ordering (FMO) [13] in conjunction with channel coding techniques. The resulting system is evaluated and is shown to outperform the recently proposed method in [5]. The performance gain is attributed to the more efficient data organization of our scheme, which allows better error concealment without sacrificing coding performance, and to the finer protection of slice groups arising from our unequal error protection strategy.

The paper is arranged as follows. The adaptive macroblock slice grouping employed by the proposed scheme is described in Section 2. Section 3 presents the proposed unequal error protection algorithm. Experimental results are reported in Section 4. Finally, conclusions are drawn in Section 5.

2. ADAPTIVE MACROBLOCK SLICE GROUPING

In this section, we present the macroblock classification policy employed by the proposed scheme. Macroblocks are rectangular picture areas and are considered the basic encoding units in H.264/AVC. Although independent encoding of macroblocks is allowed, in general, this approach is not preferable since it would require the transmission of overhead for stating the encoding parameters for each one of the independently encoded macroblocks. To overcome this problem, macroblocks are not coded as single units, but in larger groups of macroblocks, termed slices. Slices are structures of jointly encoded macroblocks which exploit spatial dependencies more effectively by partially sacrificing the error localization capabilities of the decoder. The encoding parameters of macroblocks are declared in a header (Figure 1) which includes the encoding parameters of all macroblocks in a slice. Therefore, slices are self-contained in the sense that they can be independently decoded without utilizing data from other slices of the current frame. Henceforth, each such slice will be assumed to be transmitted in a single transmission unit which will be termed "packet." The terms "packets" and "slices" will be used interchangeably in the analysis below, with "packet" meaning the transmitted stream corresponding to a slice. In this work, we assume that macroblocks are classified in three categories. This is depicted in Figure 3(a). Due to this classification, if a slice is erased, only the macroblocks which are located at slice boundaries can be concealed effectively using neighboring[1] slices that were received errorlessly at the decoder. Specifically, error-affected frame areas are efficiently concealed using the nonnormative concealment methods of [14].

The limitation of the above conventional slice formation is partially overcome in H.264/AVC, in which error concealment is improved by means of an arrangement which is termed flexible macroblock ordering (FMO). Using FMO, groups of macroblocks, known as slice groups, are formed. Slice groups consist of one or more slices; this enables better error localization. The structure of a slice group is illustrated in Figure 2. Some macroblock classification patterns, like the checkerboard (Figure 3(b)), are available in the H.264/AVC standard. As reported in [15], the FMO mode, in conjunction with advanced error concealment methods applied at the decoder, maintains the visual impact of the losses at a low level even at loss rates up to 10%, which makes it difficult for a trained eye to identify the lossy environment. Apart from predefined patterns, fully flexible macroblock ordering (explicit mode) is also allowed. According to this mode, macroblock classification into slice groups may not remain static throughout the entire video sequence, but it may change dynamically based on the video content.

The provision for dynamic formation of slice groups is exploited by the proposed system. Specifically, slice groups are formed with respect to their relative importance. As a measure of macroblock importance (based on the mean square error, MSE), we use the distortion D_{MB} defined as

$$D_{\mathrm{MB}} = \frac{1}{x_{\mathrm{MB}} \cdot y_{\mathrm{MB}}} \cdot \sum_{i=1}^{x_{\mathrm{MB}}} \sum_{j=1}^{y_{\mathrm{MB}}} \left(c_{i,j} - \widetilde{c}_{i,j}\right)^2, \tag{1}$$

where x_{MB}, y_{MB} are macroblock dimensions and $c_{i,j}$, $\widetilde{c}_{i,j}$ are, respectively, the original and the reconstructed coefficients in a macroblock. Alternatively, other metrics like the mean absolute error (MAE) could also be used.

Prior to macroblock classification, the mean value D_{mean} of the macroblock distortions is computed as

$$D_{\mathrm{mean}} = \frac{1}{N_{\mathrm{MB}}} \cdot \sum_{i=1}^{N_{\mathrm{MB}}} D_{\mathrm{MB}_i}, \tag{2}$$

where N_{MB} is the total number of macroblocks in a frame and D_{MB_i} is the distortion associated with the ith macroblock. Subsequently, the relative distortion of each macroblock is compared with D_{mean}. The macroblocks are labelled with respect to their importance as "high," "medium," and "low" as in [12]. The classification of the macroblocks into the above categories takes place using two thresholds, T_l and T_h,

[1] The term neighboring refers to both the spatial and the temporal domains. Thus, slices from the current and the previous frames are used for error concealment.

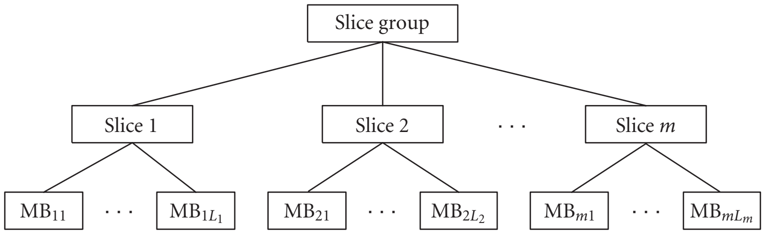


FIGURE 2: Slice group formation.

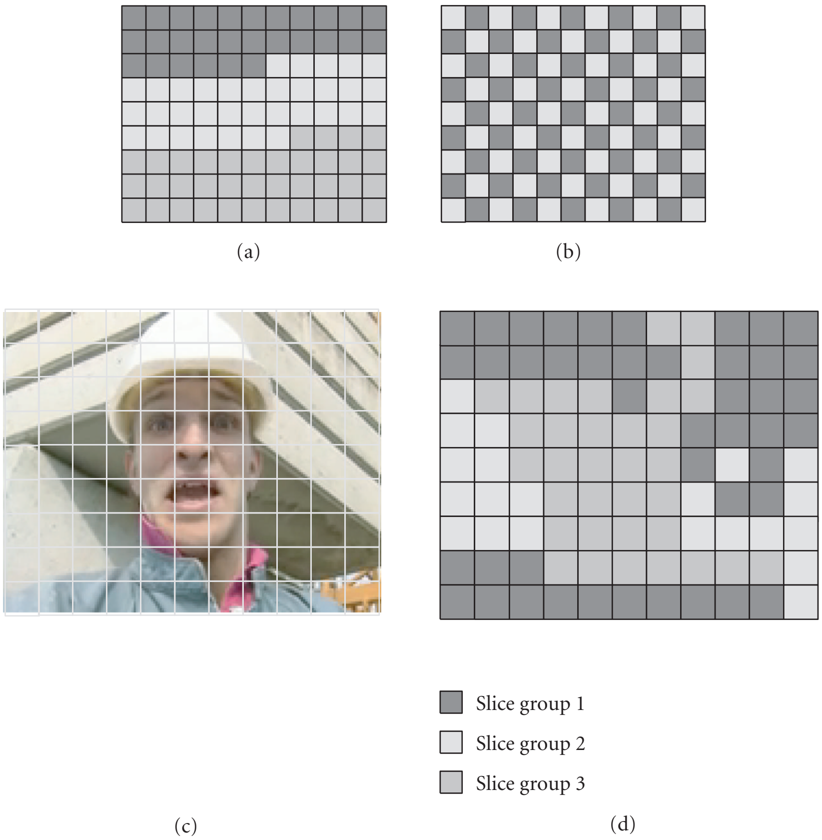


FIGURE 3: Macroblock classification (a) without FMO, (b) employing FMO (checkerboard), (c) original frame of Foreman, (d) classification map following fully FMO mode.

according to the following rules:

(i) if $D_{\text{MB}} < T_l \cdot D_{\text{mean}}$, the examined macroblock is classified to the "low" importance slice group,
(ii) if $T_l \cdot D_{\text{mean}} \leq D_{\text{MB}} < T_h \cdot D_{\text{mean}}$, the examined macroblock is classified to the "medium" importance slice group,
(iii) if $D_{\text{MB}} \geq T_h \cdot D_{\text{mean}}$, the examined macroblock is classified to the "high" importance slice group.

The distortion D_{MB} initially used is determined assuming the frame as a single slice group. After the classification of macroblocks into three slice groups, the compression efficiency will degrade and thus, more bits will be needed for the encoding of each macroblock than those initially estimated. This is taken into account by the rate-control algorithm at the encoder. In Figures 3(c) and 3(d), a frame of the Foreman sequence and its macroblock allocation map (MBAmap) for three classes, according to the above rules, are presented. The area regarded as being of high-importance mainly corresponds to intense motion or high texture regions. For example, in Figure 3(c) the "high" importance slice group coincide with foreman's head which is the main

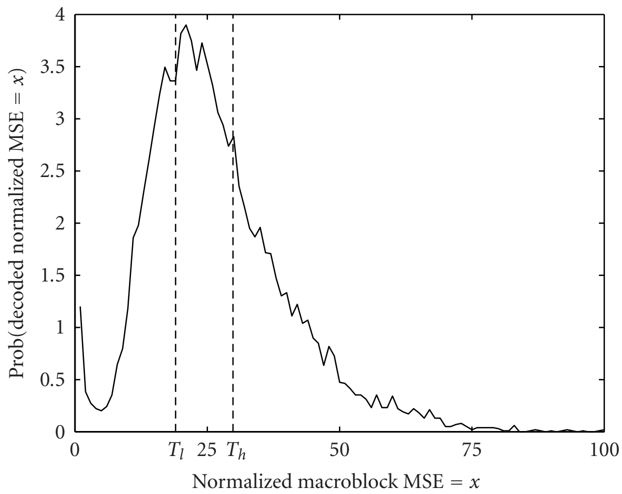


Figure 4: Histogram function of macroblocks distortion and their respective classification thresholds.

moving object in the scene, whereas the background and the body are signed as medium and low importance slice groups.

The classification of macroblocks into three categories, and not more, is reasonable, since in this way macroblocks of approximately equal importance are grouped together. Classification into more categories would not be preferable because it would lead to the generation of rather small-length packets. This is undesirable because of the increased associated packet overhead (RTP/UDP/IP overhead) containing the transmission parameters.

The determination of the thresholds T_l and T_h, which are used for the classification of macroblocks into three slice groups, will be described in Section 3. The average values of T_l and T_h are 0.7 and 1.1, respectively. It is worth noting that these threshold values are used only for the initial classification of the optimization algorithm of Section 3. These are subsequently refined during the optimization procedure. The normalized histogram function of macroblocks' distortions and the respective thresholds are illustrated in Figure 4. Following the above classification rules, slice groups are formed.

Since the transmission scenario is over packet erasure networks, channel codes should be used for the efficient protection of the H.264/AVC streams. To this end, we developed an algorithm for the efficient channel rate allocation. This is presented in the ensuing section.

3. CHANNEL RATE ALLOCATION

In the preceding analysis for an optimal classification, it was assumed that the distortion between the original and reconstructed coefficients is known. In practice, however, the actual distortion depends on the reconstructed coefficients *after* channel decoding. This means that the processes of slice grouping and channel allocation are actually interdependent. For this reason, the formation of slice groups and their unequal error protection are optimized in our system by iterating two interdependent steps.

During the channel rate allocation process, slices are transferred from one slice group to another leading to new slice group formations. The channel rate allocation algorithm classifies optimally the macroblocks into slice groups and determines their optimal channel protection. As it can be seen, the choice of the classification thresholds is an important issue. When the thresholds are close to the optimal values, the channel rate allocation procedure is made more efficient and the computational cost is significantly reduced. The thresholds used for classification at the I-frame are initially determined by experimentation and guarantee satisfactory image quality and error resiliency at the receiver. In the sequel, the thresholds are refined following an iterative technique which is described in detail below. Specifically, the resulting macroblock classification is used for the refinement of the classification thresholds. The determined thresholds are used for the initial macroblock classification in the next frame. Similarly, thresholds are determined for the remaining frames. From the above analysis, it is obvious that the FMO generates slices which can be used in conjunction with unequal error protection (UEP) schemes.

3.1. Problem formulation

Using the FMO, it is possible to form slice groups of unequal importance. In our approach, the unequally-important slice groups consist of equally sized slices (packets), that is, the size of the slices in each slice group is the same (in bytes) but the importance of the resulting slice groups is different. Therefore, UEP should be applied for their efficient protection. Reed-Solomon (RS) codes were chosen for use with our system due to their excellent error recovery properties for transmission over packet erasure networks. Since, different frames have, in general, different classification maps, channel rate allocation is performed at the frame level. The proposed algorithm takes into account the importance of each slice group and allocates more RS packets (RS slices) to slice groups car-

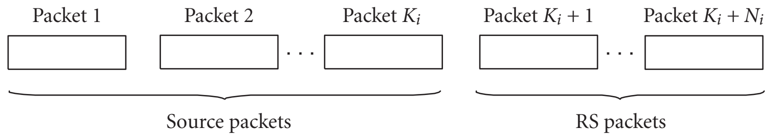


Figure 5: Packet formation of a slice group.

rying important information and less to the rest. The problem is solved optimally using dynamic programming techniques under two constraints which are presented in the following. The packet formation of a slice group after RS encoding is illustrated in Figure 5.

The distortion D_f of each frame is expressed as the sum of the individual slice group distortions $D_{f,i}$. Therefore,

$$D_f = \sum_{i=1}^{s} D_{f,i}, \tag{3}$$

where s is the number of slice groups.

The optimization objective is to find

(i) the optimal classification of macroblocks into slice groups,
(ii) the optimal RS channel protection of slice groups.

The optimization algorithm intents to minimize the average expected distortion $\overline{D}$ subject to two constraints. The first constraint is imposed by the rate control algorithm of the H.264/AVC. Hence,

$$\sum_{i=1}^{s} K_i = K_f, \tag{4}$$

where K_i is the number of source packets classified into the ith slice group of a frame, and K_f is the total number of source packets for the frame.

A channel rate constraint is required to set an upper limit to the RS protection which can be used for the protection of a frame. This reduces significantly the possible channel rate allocations and facilitates the allocation procedure. Thus, it is

$$\sum_{i=1}^{s} N_i \leq N_f, \tag{5}$$

where N_i is the number of RS packets allocated to the ith slice group and N_f is the total number of RS packets allowed for the protection of the frame.

The channel rate constraint is necessary to avoid overprotection of the first frames. Specifically, without the channel rate constraint, the first frames in the sequence would allocate the maximum allowable RS protection. Therefore, the remaining frames would have less available rate and, consequently, drift would occur. The maximum number N_f of RS packets (per frame) which can be used for the channel protection of a frame was found by experimentation. N_f is expressed as a fraction of the available source packets for each frame. In order to determine N_f and, thus, the optimal channel rate r_c of a sequence, the average expected distortion is computed for a large set of channel rates. The r_c is given by

$$r_c = \frac{\sum_{i=1}^{N_{\text{seq}}} N_{f,i} \cdot p_l}{r_T}, \tag{6}$$

where N_{seq} is the number of frames in a sequence, $N_{f,i}$ the number of RS packets in frame i, p_l the packet length, and r_T the overall transmission bit rate.

From the computed channel rates r_c, the one achieving the lowest distortion is considered as optimal. Therefore, the available bit rate for source encoding of the sequence is $r_s = (1 - r_c) \cdot r_T$.

The average expected distortion when all packets are clustered to the same slice group is defined as

$$\overline{D} = \sum_{i=1}^{N} D_f \cdot P(i) + \sum_{i=N+1}^{N+K-1} D_{f,i,1} \cdot P(i) + D_{f,PC} \cdot P(N+K), \tag{7}$$

where K, N are the number of source and channel packets, respectively, and D_f is the distortion when the number of erased packets do not exceed the allocated RS protection. $D_{f,i,1}$ (1 stands for the slice group index) is the distortion when concealment is invoked to mitigate the effect of the lost packets. $D_{f,PC}$ denotes the distortion in case all packets of the current frame are lost and frame replication follows for error concealment. In the preceding analysis, the channel rate allocation algorithm assumes that all previous frames have been received intact. Thus, no distortion is introduced due to error propagation. Although, this assumption rarely holds, in general, the resulting allocation is barely affected. Finally, $P(i)$ is the probability that i, out of $N + K$, packets are erased. It is found to be equal to

$$P(i) = \binom{N+K}{i} \cdot p^i \cdot (1-p)^{N+K-i}, \tag{8}$$

where p is the packet erasure probability associated with the channel.

We have already defined the average expected distortion when each frame is transmitted as a single slice group. Trivially, it can be proved that the expected distortion for s classes is given by

$$\begin{aligned}\overline{D} = \sum_{l=1}^{s} \Bigg\{ & \sum_{i=1}^{N_l} D_{f,l} \cdot P_l(i) + \sum_{i=N_l+1}^{N_l+K_l-1} D_{f,i,l} \\ & \cdot P_l(i) + D_{f,PC,l} \cdot P_l(N_l+K_l) \Bigg\},\end{aligned} \tag{9}$$

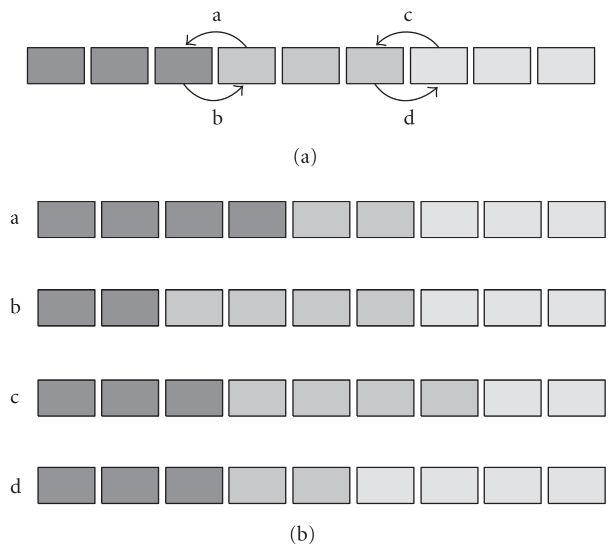


Figure 6: Allowable packet exchanges in case of three slice groups.

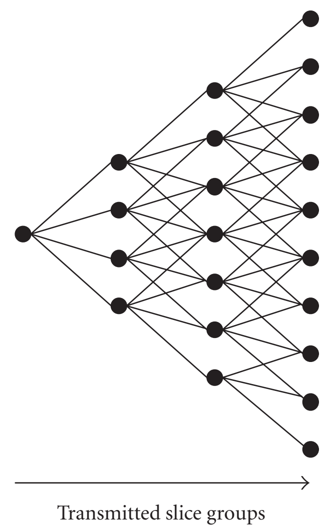


Figure 7: Trellis diagram for RS allocation.

where K_l and N_l are the number of source and RS packets of the lth slice group. $P_l(i)$ is the packet error probability of lth slice group. It is defined similar to (10) as

$$P_l(i) = \binom{N_l + K_l}{i} \cdot p^i \cdot (1-p)^{N_l+K_l-i}. \tag{10}$$

The distortion $D_{f,PC,l}$ in the last term of (9) expresses the distortion when all packets of the lth slice group are erased and concealed by slice group replication. Finally, $D_{f,i,l}$ represents the distortion introduced when the current frame slice group is concealed by slices received intact and $D_{f,l}$ the distortion when the RS protection is sufficient to recover all erased packets. It should be noted that the distortion terms do not consider error propagation. This does not affect seriously the estimated distortion since macroblocks updates usually cope effectively with drift phenomenon.

3.2. Reed-Solomon rate allocation

In this section, we present a solution to the optimization problem that was previously formulated. The optimization objective is actually two fold. Specifically, it includes the determination of both the number of slices that are classified into each slice group and their respective RS protection. In general, reaching an optimal solution of the above joint optimization problem is a difficult task. In this work, we propose a two-step optimization procedure, which iteratively determines the packet classification and the RS protection. Although, this approach to the solution of the optimization problem does not guarantee global optimization, in practice it yields very satisfactory results. The optimization procedure is summarized as follows.

(1) Determine the RS protection for each frame.
(2) Determine the thresholds T_h and T_l.
(3) Classify all macroblocks into slice groups according to T_h and T_l.
(4) Find the optimal RS protection for the above classification.
(5) Calculate the expected distortion of allowable neighboring macroblock classifications with the restriction that a single packet can be exchanged between successive classes.
(6) Compare the expected distortion of the ancestor classification with the lowest average distortion of all descendant classifications of step (3). If a classification with lower expected distortion is reached, it is considered as optimal and steps (2) to (6) are repeated, otherwise the algorithm is terminated. When the same packet is exchanged between two slice groups in two successive iterations, the algorithm is again terminated.

If three slice groups are assumed, the possible packet exchanges are illustrated in Figure 6. It is worth noting that the actual search space is limited, since only four new packet formations are possible. If a slice group does not contain any packet, the possible formations are even fewer.

Our objective is to optimize the RS allocation by minimizing the expected distortion given by (9). Although this optimization can be performed by exhaustive search among all possible channel rate allocations, this approach is not preferable since the computational cost would be prohibitive for real-time applications. However, the computational cost can be significantly reduced using the dynamic programming algorithm in [16, 17]. The trellis diagram corresponding to the minimization of (9), subject to a rate constraint, is shown in Figure 7. Each branch in the trellis corresponds to the application of a specific RS code to a slice group. The algorithm first determines the RS protection of the more important slice groups and then the respective protection of the less important slice groups. The nodes in the trellis represent the intermediate stages where decisions are made about the best RS allocation up to the sth slice group protection. Paths merging in a single node correspond to allocations that yield not only the equal source rates but also equal transmission rates. Among the paths converging to a node, the path attaining the lower expected distortion is retained (survivor) while the rest are pruned. In the final stage, among the survivor paths, the one with the lowest overall expected distortion corresponds to the optimal RS allocation. The number of states in the trellis depends on the allowable RS protection levels.

4. EXPERIMENTAL RESULTS

The proposed scheme for transmission of H.264/AVC streams over IP/UDP/RTP was evaluated using the two standard QCIF sequences Foreman and Carphone, coded at 10 frame/s (fps), and the CIF sequence Paris, coded at 30 fps. Group of pictures (GOPs) of $IPPP\ldots$ structure consisting of 100 and 300 frames were considered for the QCIF and CIF sequences, respectively. The NS-2 event simulator [18], employing a uniform bit error model, was used for channel simulations. The NS-2 was selected to simulate more realistically[2] the examined wireline transmission scenaria. It should be noted that, with minor modifications, the proposed method could also be used for wireless video transmission.

The video sequences were encoded using JM 8.3 [19] of the H.264/AVC standard [1]. The first frame in the sequence was intracoded and the following frames were intercoded. Temporal redundancy was removed using up to 1/4 pixel accuracy motion compensation. Multiple reference picture selection [20] was allowed for improved coding efficiency and error resiliency. The reference frame buffer was set to the maximum value 5. The universal variable length coding (UVLC) [1] was selected as the entropy coder. For the estimation of the end-to-end distortion, 30 independent channel-decoder pairs were used in the encoder, as suggested in [21], and nonnormative advanced error concealment methods were applied [14]. The same error concealment techniques were also applied at the decoder side.

The JM 8.3 was modified to support fully flexible macroblocks allocation map (MBAmap) for each frame. The picture parameter set (PPS) packets used by JM 8.3, which contain the classification maps, are protected using strong channel codes. Specifically, the $(3, 1)$ RS codes were used since they are able to correct all possible error patterns occurring in the considered channel conditions. The use of these RS codes is affordable because the PPS packet size is small in comparison to the average frame size. In particular, PPS packets sized 30 and 120 bytes on average for QCIF and CIF sequences, respectively, while the average frame size was between 800 and 1500 bytes for QCIF sequences and between 3000 and 6000 bytes for CIF sequences. The bit rate allocated to PPS packet protection was in the range of 5–10% of the overall transmission rate. The chosen channel coding strategy for PPS packets is needed in order to ensure that high-quality video sequences will be decodable even in the case of high packet error rates. Due to the strong protection that is applied to the PPS packets, in the sequel we assume that PPS packets are always available without errors at the decoder.

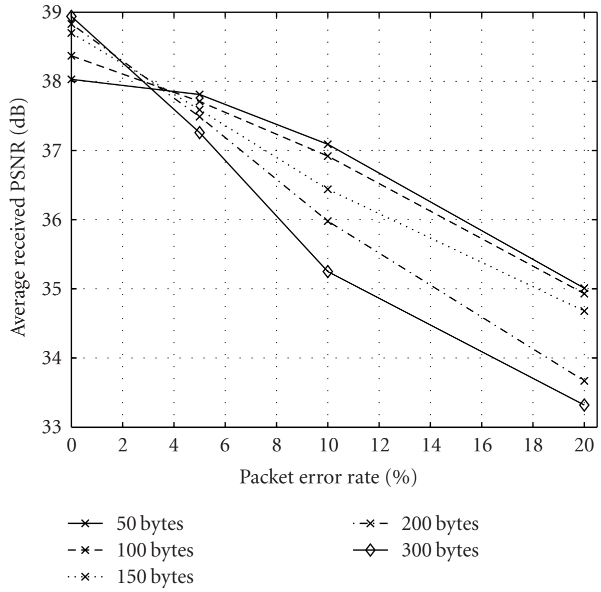


Figure 8: Average received mean PSNR for transmission of the Foreman sequence coded at 128 kbps over channels facing packet error rates in the range $[0, 20]$ for various packet sizes.

The packet sizes were 50 and 200 bytes for the QCIF and CIF sequences, respectively. The use of relatively small packet sizes endowed our scheme with the ability to achieve better error localization and prevent drift. If longer packets were used, wider frame areas would be affected in case of erasures. In such cases, errors would not be concealed effectively and the decoding process would be inefficient. The main drawback of utilizing small packets is, as expected, the less efficient compression due to the poor prediction and the increased packet overhead. This is shown in Figure 8 where it is seen that small packets guarantee the decoding of video sequences of satisfactory quality, whereas schemes with larger packets benefit in error-free cases. Considering the above, our choices of packet sizes achieve a good tradeoff between robustness and compression efficiency.

The employment of small packets could result in increased bandwidth requirements for packet headers transmission. In order to avoid this, the robust header compression (RoHC) [22] was used, which reduces the IP/UDP/RTP header from 40 bytes to approximately 3 bytes. Thus, the resulting packet overhead is about 1.5% and 6% of the overall

[2] NS-2 considers several parameters like round trip time, delay, jitter, and advanced features (e.g., drops due to congestion and bottleneck effects in concurrent flows). Although these features are not considered in our experiments, we use NS-2 for channel modelling since it is a well-known testbed and the results can be easily replicated from other researchers.

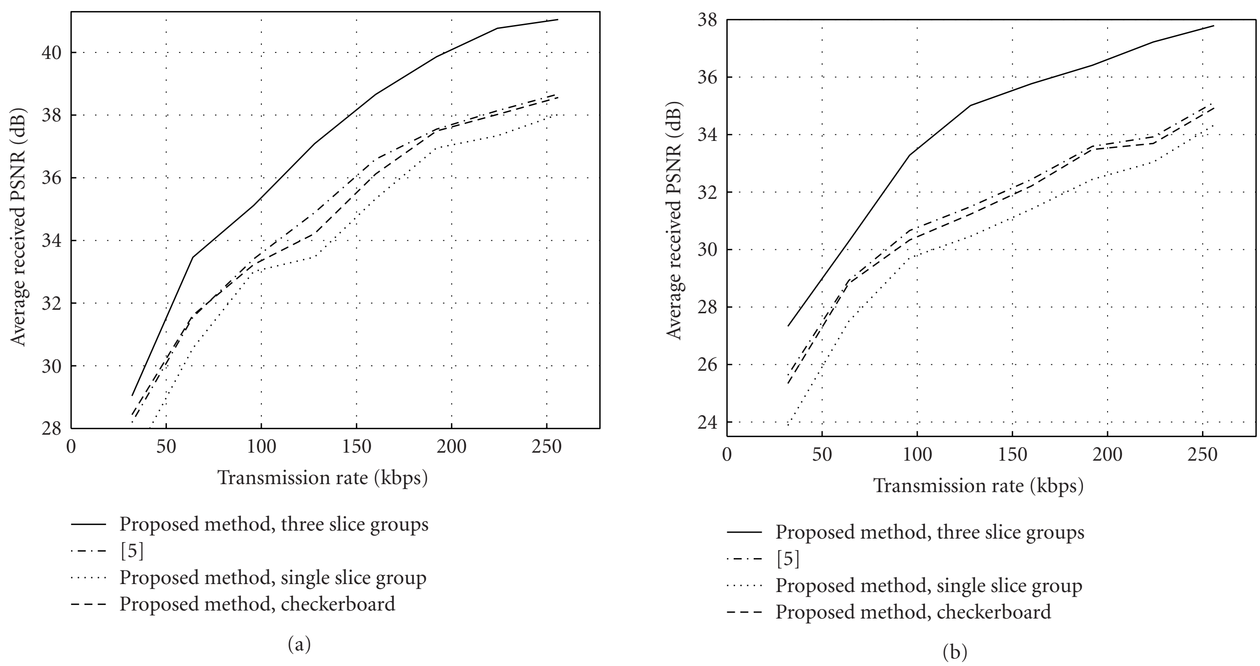


Figure 9: Comparison of the proposed methods with the method in [5] for the transmission of the QCIF sequence Foreman. Reconstruction quality in terms of mean PSNR is reported. Results for packet error rate equal (a) 10%, (b) 20%.

transmission rate for CIF and QCIF sequences, respectively. This cost is reasonable considering that small packets improve drastically the error concealment and localization capabilities of the system. The main disadvantage of RoHC is the increased processing delay at routers, which leads to end-to-end delays. However, as shown in several other techniques (e.g., in [23–27]) it is possible to use RoHC for real-time communication over multihop networks.

Adaptive slice grouping was employed by the proposed system. Specifically, as presented in Section 2, the slices were classified into three slice groups. The MSE was considered as the classification metric. Since, the slice groups are of unequal importance, different sets of RS code rates were used for their protection. Therefore, the slice groups labelled as "low" and "medium" are protected less, while stronger RS codes were used for the class of "high" importance.

Three variants of the proposed scheme were considered for comparison purposes:

(i) the full scheme, which classifies macroblocks into three slice groups according to the rules presented in Section 2,
(ii) a scheme which divides the image into two slice groups according to the checkerboard pattern,
(iii) a simplified scheme which treats each frame as a single slice group.

The RS protection for the above schemes was determined using the UEP algorithm of Section 3. Prior to channel rate allocation the optimal channel rate r_c (6) is found. Then the algorithm follows the optimization process presented in Section 3.2, which iteratively refines the estimated RS protection until a close to optimal protection is reached. From the examined RS allocations, the strongest employed RS code is the one which allocates all RS packets to the most important slice group. In particular, if K_i is the number of source packets of the ith slice group, then the examined RS codes are part of the $(K_i + \xi, K_i)$ family, where $\xi \in [0, N_f]$.[3]

The peak-signal-to-noise ratio (PSNR) was used as a measure of the reconstruction quality. As in almost all related literature, in the present work we report results in terms of mean PSNR. All reported results are averages over 100 simulations. The proposed schemes are compared with an implementation of the method in [5] which uses two data partitions and employs slices of fixed number of macroblocks. The optimization of [5] was applied at the NAL level. The method in [5] was selected for comparison purposes since it is a joint source/channel coding scheme which is in the spirit of our method. The transmission schemes were evaluated for a variety of channel conditions. In Figures 9(a), 10(a), and 11(a), results for transmission over packet networks with 10% packet losses are presented for the Foreman, Carphone, and Paris video sequences. Optimization was performed assuming 10% packet error rate. From Figures 9(a), 10(a), and 11(a), it can be easily seen that the three slice group variant of the proposed method decodes higher-quality videos more frequently than the rest of the methods. The performance gap between our best-performing scheme and the method in [5] is significant and grows wider as the transmission bit rate

[3] Typical values for K_i and N_f range from 3 to 10 and from 0 to 10, respectively.

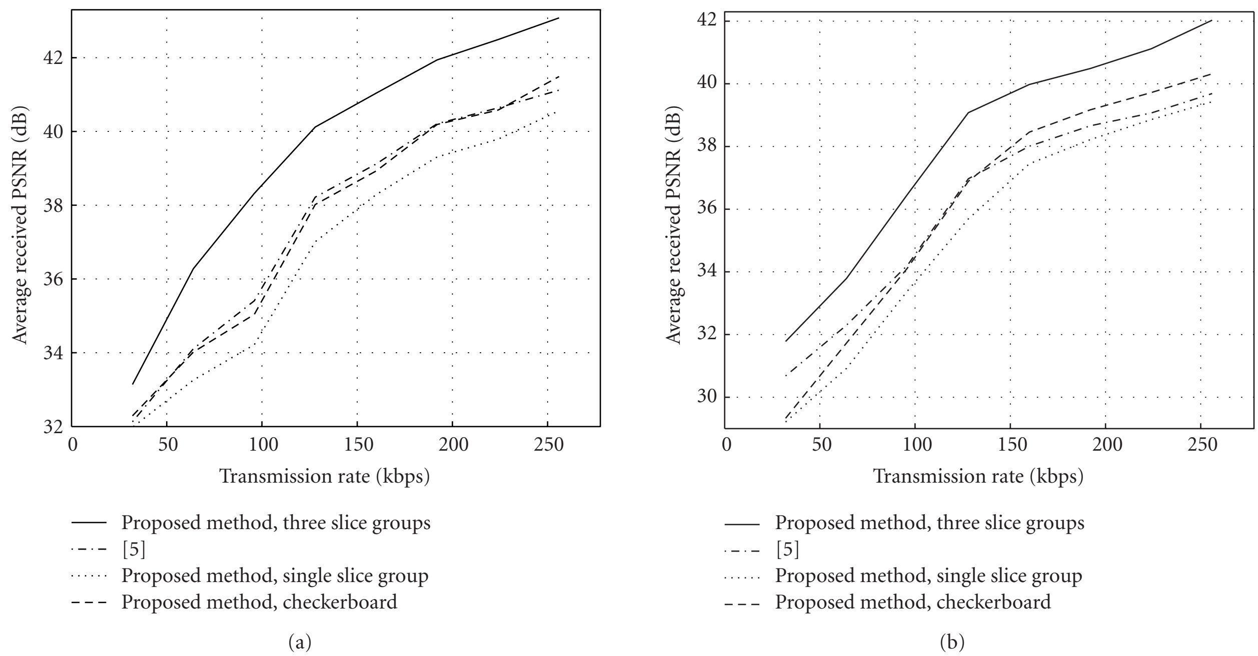


Figure 10: Comparison of the proposed methods with the method in [5] for the transmission of the QCIF sequence Carphone. Reconstruction quality in terms of mean PSNR is reported. Results for packet error rate equal (a) 10%, (b) 20%.

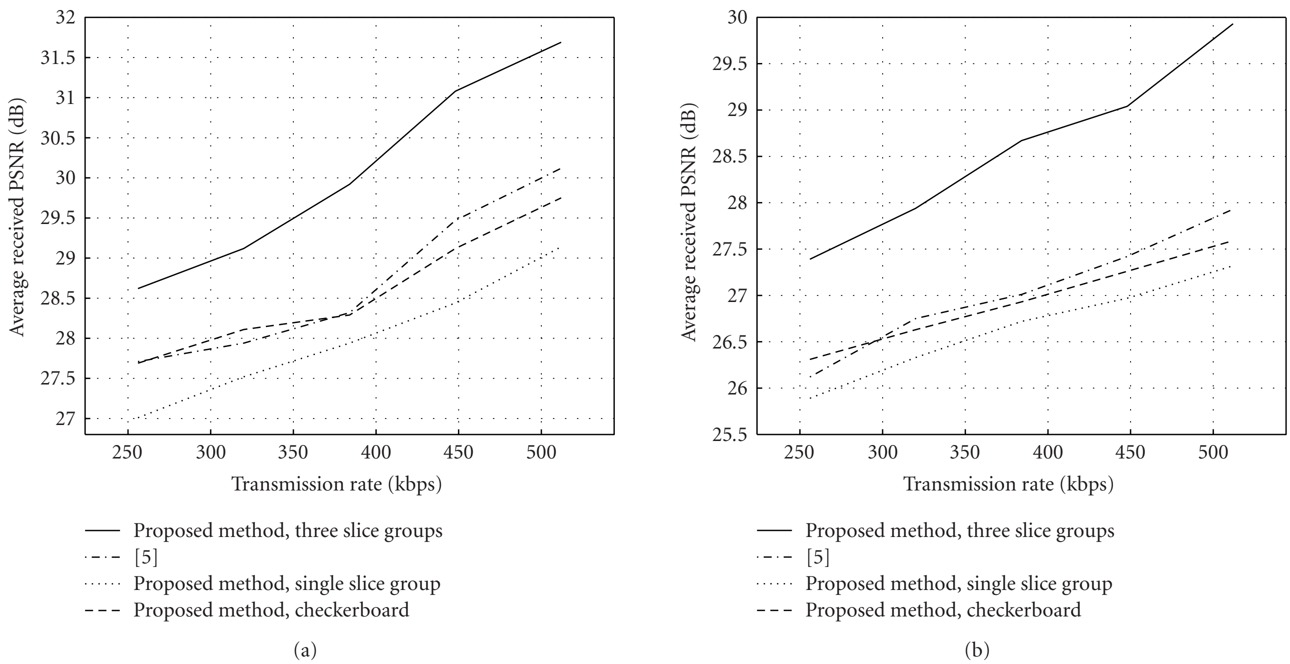


Figure 11: Comparison of the proposed methods with the method in [5] for the transmission of the CIF sequence Paris. Reconstruction quality in terms of mean PSNR is reported. Results for packet error rate equal (a) 10%, (b) 20%.

increases. The performance gains achieved using the proposed scheme is due to the adaptive slice grouping which enables better error localization as well as the efficient error protection. From Figures 9, 10, and 11 it is obvious that our three slice group approach performs significantly better than other variants of our scheme (i.e., single-sliced scheme). The unequal error protection algorithm also boosts the performance of the proposed scheme, since the unequal protection

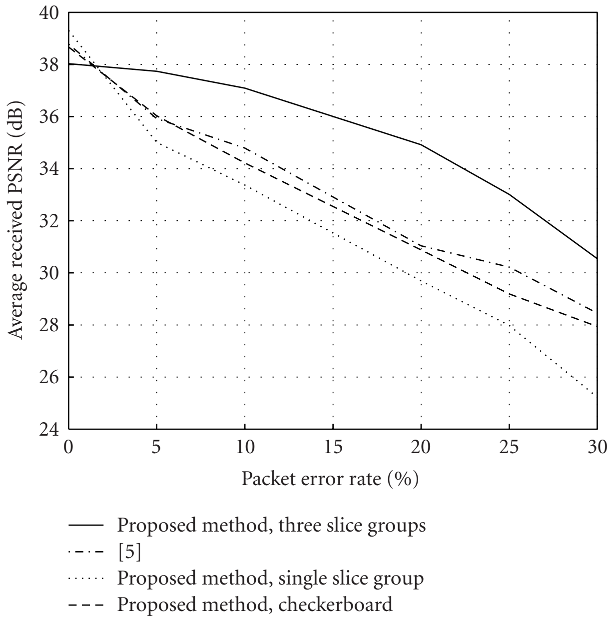


Figure 12: PSNR comparison for the transmission of the QCIF sequence Foreman at 128 kbps as a function of the packet error rate. The scheme was optimized for 10% packet error rate and tested for various packet error rates.

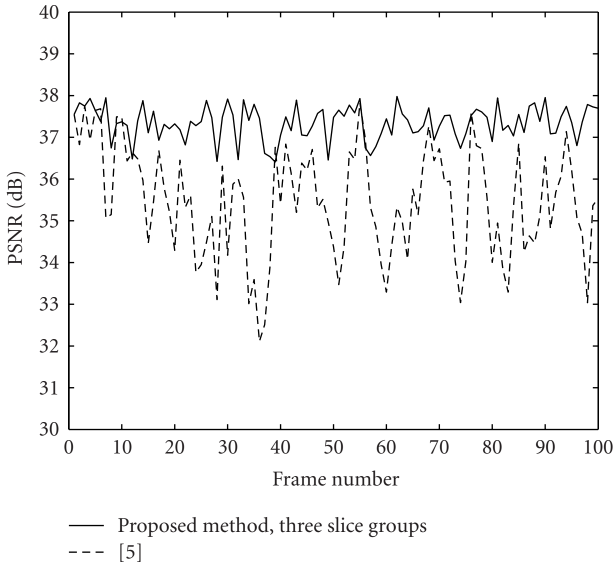


Figure 13: PSNR comparison of the proposed full scheme with the method in [5] for the transmission of the QCIF sequence Foreman coded at 128 kbps over packet erasure channel with 10% packet losses.

of slice groups enables the application of less powerful RS codes, and thus, saves rate which can be used for the transmission of source rate. Considering the above, the performance gain should not be attributed solely to the adaptive group slicing itself or the UEP algorithm, but rather to their synergistic cooperation.

Transmission of video over more unreliable channels was also considered. The schemes were optimized for 20% packet error rate and transmitted over packet erasure networks which encounter the considered channel conditions. For the Foreman, Carphone, and Paris sequences the results are presented in Figures 9(b), 10(b), and 11(b), respectively. The results clearly and consistently demonstrate the superiority of the proposed scheme with multiple slice groups and verify the conclusions reached for less noisy channels. As previously, the performance gain stems from both the slice group classification and the optimal channel rate allocation algorithm.

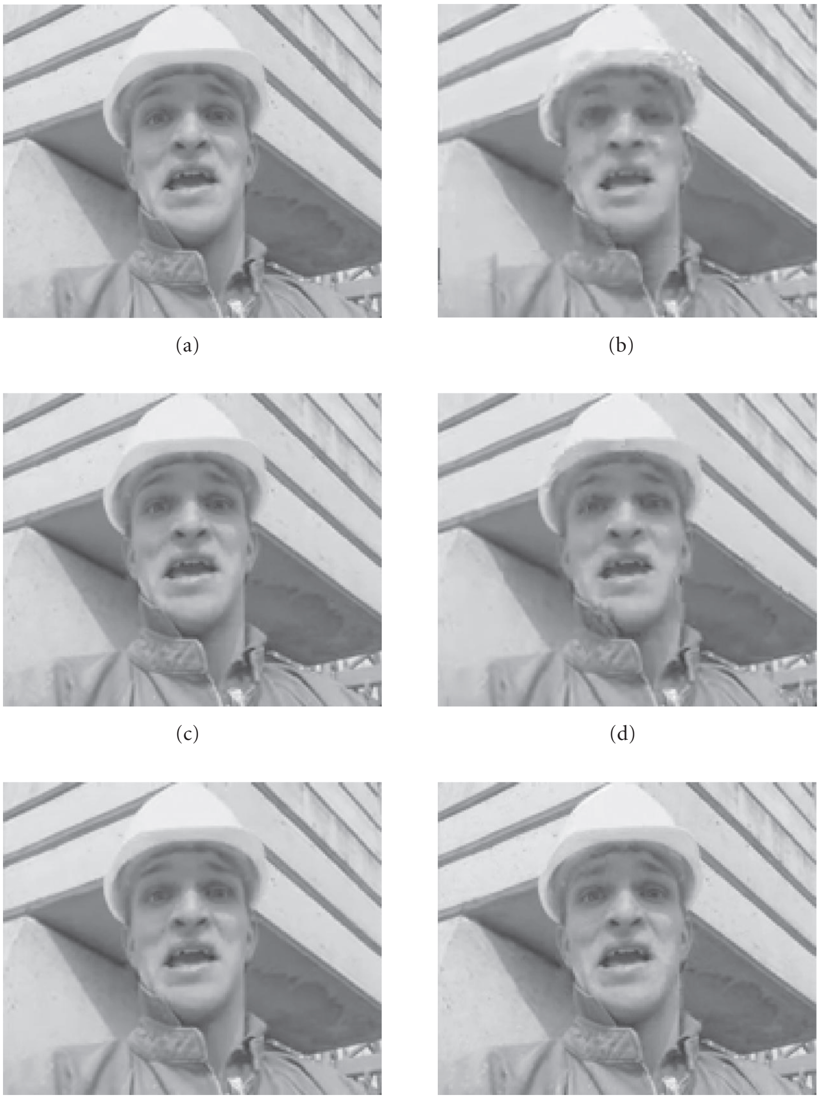
The proposed scheme was also evaluated for transmission in channel mismatch conditions. In Figure 12, results are presented for Foreman QCIF sequence coded at 128 kbps for the case where the schemes are optimized for packet error rate equal to 10% and transmitted over channels which exhibit various packet error rates. The results show that the proposed full scheme is superior to the method in [5] and the other variants of the full scheme. When the transmission is error free, the proposed full scheme has lower performance due to the application of stronger RS codes and the inferior compression efficiency when FMO is used. The gain achieved by the full scheme over the other methods becomes more impressive when the channel conditions deteriorate. Specifically, for the most of the considered transmission scenarios, the performance gap is roughly 2 dB. It is worth noting that our three slice group method provides graceful degradation in image quality when the channel becomes noisier, whereas the other methods collapse. This is due to the exploitation of adaptive slice grouping which improves the performance of error concealment methods and the channel rate allocation algorithm of Section 3.

For the sake of the comparison, in Figure 13 the full scheme is compared, in terms of PSNR, with the method in [5] for transmission of Foreman over channel with 10% packet losses. As it can be seen, the proposed scheme is, in general, more robust to packet losses. Moreover, the reconstruction quality degrades more gracefully. On the contrary, the method in [5] exhibits unpleasant fluctuations in image quality.

In Figure 14, we present a visual comparison of the decoded sequences by the proposed methods. From Figure 14, we can see that the three slice group variant of the proposed method outperforms the other variants. It should also be noticed that the proposed method does not induce annoying artifacts.

5. CONCLUSIONS

A novel method was proposed for the transmission of H.264/AVC-coded sequences over packet erasure channels. The proposed scheme exploits the error resilient features of H.264/AVC codec and employs Reed-Solomon codes to protect effectively the resulting streams. A novel macroblock classification scheme into three slice groups was used for



(a) (b) (c) (d) (e) (f)

Figure 14: Visual comparison of the proposed methods using the frame 68 of the Foreman sequence coded at 96 kbps. Comparison of visual artifacts induced due to transmission over packet networks encountering 10% packet error rate. Error-free transmission of the (a) single slice group variant of the proposed scheme (37.32 dB), (c) two slice groups (checkerboard) variant of the proposed scheme (36.58 dB), (e) three slice groups variant of the proposed scheme (36.06 dB). Frames harmed by noise when sequences are encoded using the (b) single slice group variant of the proposed scheme (32.86 dB), (d) two slice groups (checkerboard) variant of the proposed scheme (33.47 dB), (f) three slice groups variant of the proposed scheme (34.93 dB).

improved error resilience. A framework for optimal classification of macroblocks into slice groups and optimal unequal error protection was also proposed. Experimental evaluation showed the superiority of the proposed method in comparison to well-known schemes for transmission of H.264/AVC streams.

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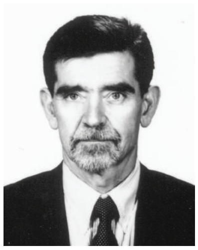

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Transforming Signal Processing Applications into Parallel Implementations

Call for Papers

There is an increasing need to develop efficient "system-level" models, methods, and tools to support designers to quickly transform signal processing application specification to heterogeneous hardware and software architectures such as arrays of DSPs, heterogeneous platforms involving microprocessors, DSPs and FPGAs, and other evolving multiprocessor SoC architectures. Typically, the design process involves aspects of application and architecture modeling as well as transformations to translate the application models to architecture models for subsequent performance analysis and design space exploration. Accurate predictions are indispensable because next generation signal processing applications, for example, audio, video, and array signal processing impose high throughput, real-time and energy constraints that can no longer be served by a single DSP.

There are a number of key issues in transforming application models into parallel implementations that are not addressed in current approaches. These are engineering the application specification, transforming application specification, or representation of the architecture specification as well as communication models such as data transfer and synchronization primitives in both models.

The purpose of this call for papers is to address approaches that include application transformations in the performance, analysis, and design space exploration efforts when taking signal processing applications to concurrent and parallel implementations. The Guest Editors are soliciting contributions in joint application and architecture space exploration that outperform the current architecture-only design space exploration methods and tools.

Topics of interest for this special issue include but are not limited to:

- modeling applications in terms of (abstract) control-dataflow graph, dataflow graph, and process network models of computation (MoC)
- transforming application models or algorithmic engineering
- transforming application MoCs to architecture MoCs
- joint application and architecture space exploration
- joint application and architecture performance analysis
- extending the concept of algorithmic engineering to architecture engineering
- design cases and applications mapped on multiprocessor, homogeneous, or heterogeneous SOCs, showing joint optimization of application and architecture

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Special Issue on Video Adaptation for Heterogeneous Environments

Call for Papers

The explosive growth of compressed video streams and repositories accessible worldwide, the recent addition of new video-related standards such as H.264/AVC, MPEG-7, and MPEG-21, and the ever-increasing prevalence of heterogeneous, video-enabled terminals such as computer, TV, mobile phones, and personal digital assistants have escalated the need for efficient and effective techniques for adapting compressed videos to better suit the different capabilities, constraints, and requirements of various transmission networks, applications, and end users. For instance, Universal Multimedia Access (UMA) advocates the provision and adaptation of the same multimedia content for different networks, terminals, and user preferences.

Video adaptation is an emerging field that offers a rich body of knowledge and techniques for handling the huge variation of resource constraints (e.g., bandwidth, display capability, processing speed, and power consumption) and the large diversity of user tasks in pervasive media applications. Considerable amounts of research and development activities in industry and academia have been devoted to answering the many challenges in making better use of video content across systems and applications of various kinds.

Video adaptation may apply to individual or multiple video streams and may call for different means depending on the objectives and requirements of adaptation. Transcoding, transmoding (cross-modality transcoding), scalable content representation, content abstraction and summarization are popular means for video adaptation. In addition, video content analysis and understanding, including low-level feature analysis and high-level semantics understanding, play an important role in video adaptation as essential video content can be better preserved.

The aim of this special issue is to present state-of-the-art developments in this flourishing and important research field. Contributions in theoretical study, architecture design, performance analysis, complexity reduction, and real-world applications are all welcome.

Topics of interest include (but are not limited to):

- Heterogeneous video transcoding
- Scalable video coding
- Dynamic bitstream switching for video adaptation
- Signal, structural, and semantic-level video adaptation
- Content analysis and understanding for video adaptation
- Video summarization and abstraction
- Copyright protection for video adaptation
- Crossmedia techniques for video adaptation
- Testing, field trials, and applications of video adaptation services
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Special Issue on

Knowledge-Assisted Media Analysis for Interactive Multimedia Applications

Call for Papers

It is broadly acknowledged that the development of enabling technologies for new forms of interactive multimedia services requires a targeted confluence of knowledge, semantics, and low-level media processing. The convergence of these areas is key to many applications including interactive TV, networked medical imaging, vision-based surveillance and multimedia visualization, navigation, search, and retrieval. The latter is a crucial application since the exponential growth of audiovisual data, along with the critical lack of tools to record the data in a well-structured form, is rendering useless vast portions of available content. To overcome this problem, there is need for technology that is able to produce accurate levels of abstraction in order to annotate and retrieve content using queries that are natural to humans. Such technology will help narrow the gap between low-level features or content descriptors that can be computed automatically, and the richness and subjectivity of semantics in user queries and high-level human interpretations of audiovisual media.

This special issue focuses on truly integrative research targeting of what can be disparate disciplines including image processing, knowledge engineering, information retrieval, semantic, analysis, and artificial intelligence. High-quality and novel contributions addressing theoretical and practical aspects are solicited. Specifically, the following topics are of interest:

- Semantics-based multimedia analysis
- Context-based multimedia mining
- Intelligent exploitation of user relevance feedback
- Knowledge acquisition from multimedia contents
- Semantics based interaction with multimedia
- Integration of multimedia processing and Semantic Web technologies to enable automatic content sharing, processing, and interpretation
- Content, user, and network aware media engineering
- Multimodal techniques, high-dimensionality reduction, and low level feature fusion

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Special Issue on Super-resolution Enhancement of Digital Video

Call for Papers

When designing a system for image acquisition, there is generally a desire for high spatial resolution and a wide field-of-view. To achieve this, a camera system must typically employ small f-number optics. This produces an image with very high spatial-frequency bandwidth at the focal plane. To avoid aliasing caused by undersampling, the corresponding focal plane array (FPA) must be sufficiently dense. However, cost and fabrication complexities may make this impractical. More fundamentally, smaller detectors capture fewer photons, which can lead to potentially severe noise levels in the acquired imagery. Considering these factors, one may choose to accept a certain level of undersampling or to sacrifice some optical resolution and/or field-of-view.

In image super-resolution (SR), postprocessing is used to obtain images with resolutions that go beyond the conventional limits of the uncompensated imaging system. In some systems, the primary limiting factor is the optical resolution of the image in the focal plane as defined by the cut-off frequency of the optics. We use the term "optical SR" to refer to SR methods that aim to create an image with valid spatial-frequency content that goes beyond the cut-off frequency of the optics. Such techniques typically must rely on extensive a priori information. In other image acquisition systems, the limiting factor may be the density of the FPA, subsequent postprocessing requirements, or transmission bitrate constraints that require data compression. We refer to the process of overcoming the limitations of the FPA in order to obtain the full resolution afforded by the selected optics as "detector SR." Note that some methods may seek to perform both optical and detector SR.

Detector SR algorithms generally process a set of low-resolution aliased frames from a video sequence to produce a high-resolution frame. When subpixel relative motion is present between the objects in the scene and the detector array, a unique set of scene samples are acquired for each frame. This provides the mechanism for effectively increasing the spatial sampling rate of the imaging system without reducing the physical size of the detectors.

With increasing interest in surveillance and the proliferation of digital imaging and video, SR has become a rapidly growing field. Recent advances in SR include innovative algorithms, generalized methods, real-time implementations, and novel applications. The purpose of this special issue is to present leading research and development in the area of super-resolution for digital video. Topics of interest for this special issue include but are not limited to:

- Detector and optical SR algorithms for video
- Real-time or near-real-time SR implementations
- Innovative color SR processing
- Novel SR applications such as improved object detection, recognition, and tracking
- Super-resolution from compressed video
- Subpixel image registration and optical flow

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Special Issue on

Advanced Signal Processing and Computational Intelligence Techniques for Power Line Communications

Call for Papers

In recent years, increased demand for fast Internet access and new multimedia services, the development of new and feasible signal processing techniques associated with faster and low-cost digital signal processors, as well as the deregulation of the telecommunications market have placed major emphasis on the value of investigating hostile media, such as powerline (PL) channels for high-rate data transmissions.

Nowadays, some companies are offering powerline communications (PLC) modems with mean and peak bit-rates around 100 Mbps and 200 Mbps, respectively. However, advanced broadband powerline communications (BPLC) modems will surpass this performance. For accomplishing it, some special schemes or solutions for coping with the following issues should be addressed: (i) considerable differences between powerline network topologies; (ii) hostile properties of PL channels, such as attenuation proportional to high frequencies and long distances, high-power impulse noise occurrences, time-varying behavior, and strong inter-symbol interference (ISI) effects; (iv) electromagnetic compatibility with other well-established communication systems working in the same spectrum, (v) climatic conditions in different parts of the world; (vii) reliability and QoS guarantee for video and voice transmissions; and (vi) different demands and needs from developed, developing, and poor countries.

These issues can lead to exciting research frontiers with very promising results if signal processing, digital communication, and computational intelligence techniques are effectively and efficiently combined.

The goal of this special issue is to introduce signal processing, digital communication, and computational intelligence tools either individually or in combined form for advancing reliable and powerful future generations of powerline communication solutions that can be suited with for applications in developed, developing, and poor countries.

Topics of interest include (but are not limited to)

- Multicarrier, spread spectrum, and single carrier techniques
- Channel modeling
- Channel coding and equalization techniques
- Multiuser detection and multiple access techniques
- Synchronization techniques
- Impulse noise cancellation techniques
- FPGA, ASIC, and DSP implementation issues of PLC modems
- Error resilience, error concealment, and Joint source-channel design methods for video transmission through PL channels

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Numerical Linear Algebra in Signal Processing Applications

Call for Papers

The cross-fertilization between numerical linear algebra and digital signal processing has been very fruitful in the last decades. The interaction between them has been growing, leading to many new algorithms.

Numerical linear algebra tools, such as eigenvalue and singular value decomposition and their higher-extension, least squares, total least squares, recursive least squares, regularization, orthogonality, and projections, are the kernels of powerful and numerically robust algorithms.

The goal of this special issue is to present new efficient and reliable numerical linear algebra tools for signal processing applications. Areas and topics of interest for this special issue include (but are not limited to):

- Singular value and eigenvalue decompositions, including applications.
- Fourier, Toeplitz, Cauchy, Vandermonde and semi-separable matrices, including special algorithms and architectures.
- Recursive least squares in digital signal processing.
- Updating and downdating techniques in linear algebra and signal processing.
- Stability and sensitivity analysis of special recursive least-squares problems.
- Numerical linear algebra in:
  - Biomedical signal processing applications.
  - Adaptive filters.
  - Remote sensing.
  - Acoustic echo cancellation.
  - Blind signal separation and multiuser detection.
  - Multidimensional harmonic retrieval and direction-of-arrival estimation.
  - Applications in wireless communications.
  - Applications in pattern analysis and statistical modeling.
  - Sensor array processing.

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Special Issue on

Wavelets in Source Coding, Communications, and Networks

Call for Papers

Wavelet transforms are arguably the most powerful, and most widely-used, tool to arise in the field of signal processing in the last several decades. Their inherent capacity for multiresolution representation akin to the operation of the human visual system motivated a quick adoption and widespread use of wavelets in image-processing applications. Indeed, wavelet-based algorithms have dominated image compression for over a decade, and wavelet-based source coding is now emerging in other domains. For example, recent wavelet-based video coders exploit techniques such as motion-compensated temporal filtering to yield effective video compression with full temporal, spatial, and fidelity scalability. Additionally, wavelets are increasingly used in the source coding of remote-sensing, satellite, and other geospatial imagery. Furthermore, wavelets are starting to be deployed beyond the source-coding realm with increas ing-interest in robust communication of images and video over both wired and wireless networks. In particular, wavelets have been recently proposed for joint source-channel coding and multiple-description coding. This special issue will explore these and other latest advances in the theory and application of wavelets.

Specifically, this special issue will gather high-quality, original contributions on all aspects of the application of wavelets and wavelet theory to source coding, communications, and network transmission of images and video. Topics of interest include (but are not limited to) the theory and applications of wavelets in:

- Scalable image and video coding
- Motion-compensated temporal filtering
- Source coding of images and video via frames and overcomplete representations
- Geometric and adaptive multiresolution image and video representations
- Multiple-description coding of images and video
- Joint source-channel coding of images and video
- Distributed source coding of images and video
- Robust coding of images and video for wired and wireless packet networks
- Network adaption and transcoding of images and video
- Coding and communication of images and video in sensor networks

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